2006

Scalable delivery of immersive communication environment (ICE) using peer-to-peer (P2P) architecture

Ying Peng Que
University of Wollongong, ypq01@uow.edu.au

P. Boustead
University of Wollongong, boustead@uow.edu.au

Farzad Safaei
University of Wollongong, farzad@uow.edu.au

Scalable delivery of immersive communication environment (ICE) using peer-to-peer (P2P) architecture

Abstract

In this work, we propose a peer-to-peer (P2P) voice delivery architecture for the support of an immersive communication environment (ICE). An ICE service allows multiple users in a distributed virtual environment (DVE) to exchange live voices which are rendered with the directional and distance cues corresponding to the users' positions in the DVE. Our architecture addresses the scalability issue in peer access bandwidth consumption encountered by the "brute force" full- mesh P2P architecture. On the other hand, our P2P architecture seeks a good balance between access bandwidth scalability and the voice quality delivered. The three key aspects of voice quality considered are voice transmission delay, directional voice rendering accuracy and voice attenuation accuracy. Our P2P architecture achieves access bandwidth reduction by organising all the pair-wise close peers into non-overlapping clusters and by mixing the voice streams within the same cluster so as to reduce the number of direct peer to peer voice exchanges. Different sets of experiments were conducted to compare the performances of our P2P architecture against the benchmark of the full-mesh P2P architecture.

Disciplines

Physical Sciences and Mathematics

Publication Details

Que, Y., Boustead, P. A. & Safaei, F. (2006). Scalable delivery of immersive communication environment (ICE) using peer-to-peer (P2P) architecture. IEEE Global Telecommunications Conference (pp. 1-5). USA: IEEE.

Scalable Delivery of Immersive Communication Environment (ICE) using Peer-to-Peer (P2P) Architecture

Ying Peng Que, Paul Boustead, Farzad Safaei
Smart Internet CRC,
Telecommunications and Information Technology Research Institute,
University of Wollongong, Australia
Email {ying, paul, farzad}@titr.uow.edu.au

***Abstract*-In this work, we propose a Peer-to-Peer (P2P) voice delivery architecture for the support of an Immersive Communication Environment (ICE). An ICE service allows multiple users in a Distributed Virtual Environment (DVE) to exchange live voices which are rendered with the directional and distance cues corresponding to the users' positions in the DVE. Our architecture addresses the scalability issue in peer access bandwidth consumption encountered by the *"brute force"* full-mesh P2P architecture. On the other hand, our P2P architecture seeks a good balance between access bandwidth scalability and the voice quality delivered. The three key aspects of voice quality considered are voice transmission delay, directional voice rendering accuracy and voice attenuation accuracy. Our P2P architecture achieves access bandwidth reduction by organising all the pair-wise close peers into non-overlapping clusters and by mixing the voice streams within the same cluster so as to reduce the number of direct peer to peer voice exchanges. Different sets of experiments were conducted to compare the performances of our P2P architecture against the benchmark of the full-mesh P2P architecture.**

Keywords: VoIP Service for Online Gaming, Peer_to_peer Network Services for Multimedia Communications

I. Introduction

Recently, networked voice communication services in Distributed Virtual Environments (DVE) have attracted much research interest [1] [2] [3]. One typical example of DVE is Multi-player Online Games (MOG) such as Lineage II which had 2.1 million subscribers in January, 2005 [4]. Each DVE user is represented by an avatar in the virtual world. In the context of this work, there is a one to one correspondence between an avatar in the virtual world and a peer platform in the physical network. For clarity, we use the word peer only. The close interactions and co-operations between avatars in a DVE are likely to be improved with the addition of a multi-party immersive voice communication service, which immerses each avatar in a personalised *auditory scene*. The auditory scene for a particular avatar is a mixture of the voice streams from all the surrounding avatars in that avatar's hearing range. Each of these voice streams is directionally rendered and distance-attenuated according to the corresponding speaking avatar's position in the DVE. We refer to such a multi-party immersive voice communication service as an Immersive Communication Environment (ICE). Most of the Voice-over-IP (VoIP) systems currently in use with DVE deliver a party-line mixture of voice streams which contains no directional or distance cues as offered by ICE [5] [6] [7]. Our ICE system is transparent to the underlying Virtual World Applications. ICE can be either an add-on service to a Game State Server or incorporated as part of the Game Engine, e.g. the text messaging service in World of Warcrafts. ICE can be applied to any applications with avatars moving in a virtual world with defined coordinates.

Most of the aforementioned VoIP systems used for DVE are supported by the full-mesh P2P architecture. In a *full-mesh* P2P architecture, all the communicating peers exchange voice streams directly with each other. Hence, the *full-mesh* P2P architecture offers the best voice transmission delay performance [1]. However, when implementing the *full-mesh* P2P architecture using multiple uni-cast flows, on average, each peer must download and upload one voice stream from each peer in its hearing range. Therefore the access bandwidth consumption at each peer scales up linearly with the peer density D which is the average density of each peer's hearing range. Skype [6] adopts an architecture in which the peer clients with more processing and access bandwidth capacity automatically become *super peers*. Each *super peer* acts as a mixing node for all other peers in its hearing range. Skype reduces the demands on the access bandwidth of some peers but at the expense of the *super* peers. Consequently, Skype does not reduce the total number of voice exchanges required. Our proposed P2P architecture aims to address the access bandwidth scalability problem inherent in the *full-mesh* P2P architecture. Our P2P architecture clusters together pair-wise close (by virtual world distance) peers and mixes the co-cluster peer voice streams, thus reducing the number of direct peer to peer voice exchanges. Our cluster-based P2P architecture seeks a good balance between system scalability in terms of access bandwidth consumptions and the voice qualities delivered. The key voice quality requirements addressed in this work are, the voice transmission delays, directional voice rendering error and voice distance attenuation error. Moreover, in the actual routing of voice streams, we consider the important issue of even distribution of individual access bandwidth consumptions (nodal stress) between the peers.

II. Cluster Formation

A. Cluster Formulation

Due to the large complexity in an optimal formulation (at least a quadratic linear programming problem), we devised two formulations which work in tandem together to provide a sub-optimal yet more scalable solution to the problem of peer clustering. In the first stage, the *Initial Cluster Formation Formulation* finds the largest *initial cluster* around each peer (N initial clusters in total) in which the virtual world distances between all the pairs of peers are within the pre-defined *pair-wise distance constraint*. We always set the *pair-wise*

1-4244-0357-X/06/$20.00 ©2006 IEEE
This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE GLOBECOM 2006 proceedings.

distance constraint to be less than the maximal hearing range of a peer so that all the peers in the same cluster can hear each other. In the second stage, the *Final Cluster Formation Formulation* chooses to establish a subset of *final clusters* from the *N initial clusters* provided by the first stage. In the second stage of clustering, each peer is assigned to just one *final cluster*.

B. Inter-Cluster and Intra-Cluster Voice Exchanges and Renderings

There is no distance attenuation error introduced within the cluster. We define the directional voice rendering error using the concept of *acceptable voice rendering error* described in [8]. High quality voice streams are delivered to the peers residing in the same cluster because they are close to each other as bounded by the *pair-wise distance constraint.* However, not all the peers located close to each other are enclosed in the same cluster. In Fig. 1, peers P_4 and P_5 are close to P_1 (within the *pair-wise distance constraint*) but are not enclosed in C_1 with P_1. This could be because during the *initial cluster formation*, P_4 and P_5 are found to be not close enough to other peers in C_1. Alternatively, P_4 and P_5 could have been in the same *initial cluster* as P_1 but they were not assigned to the same *final cluster* as P_1 due to the need to balance between cluster size and the number of clusters (see A in Section V).

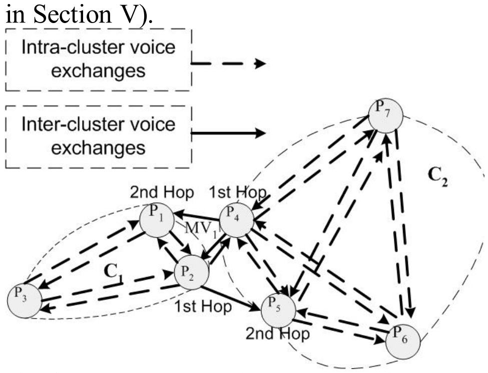

Fig. 1 Inter-cluster and intra-cluster voice exchanges and minimal delay routing

At the completion of the final cluster formation process, the peers heard by a given peer P_j are enclosed in one or more surrounding clusters. In turn, P_j receives one mixed voice stream from each of these surrounding clusters. Hence, our clustering model replaces some of the full mesh voice exchanges performed at each peer with the bandwidth-efficient inter-cluster voice exchanges. In the scenario depicted by Fig.1, peer P_1 in cluster C_1 receives one mixed voice stream MV_1 (mixture of the voices of P_4 and P_5) from the neighbouring cluster C_2 instead of having to receive one each from P_4 and P_5 as in the full-mesh P2P case. Directionally, a given mixed voice stream is rendered at a *centroid* position which offers the minimal sum of *percentage directional error*. The *percentage directional error* is the percentage difference between the *actual angular deviations* of all the peer voices rendered and their corresponding *acceptable angular errors* constraints [8]. The *attenuation error* occurs when a given listening peer receives a voice stream that is not attenuated to the exact distance between the listening peer and the corresponding speaking peer. The distance attenuation of the mixed voice stream received by a particular peer P_j from a given cluster C is governed by the minimal distance between P_j and C. This minimal distance is the shortest distance from P_j to one of the peers in C which is heard by P_j.

III. Inter-cluster Routing Algorthms

These two routing algorithms solve the problem of routing voice streams between clusters. These two routing algorithms always adhere to the principle of each peer only receiving one stream from each of the clusters that peer hears.

A. Minimal Delay Routing Algorithm

The *Minimal Delay Routing* algorithm finds the shortest path (measured by absolute delay) between any given peer and a cluster which has at least one speaking peers heard by that listening peer. For instance, in Fig. 1, the *Minimal Delay Routing* algorithm routes the voice streams between P_1 and C_2 along the shortest path starting at P_4 which offers the shortest paths between P_1 and all the peers in C_2 heard by P_1. The Minimal Delay Routing Algorithm creates a problem of uneven distribution of inter-cluster voice uploads at the peers. Some peers perform more inter-cluster voice uploads than inter-cluster voice downloads whereas others do not perform any voice uploads. In Table 1, the second and third columns from the left (marked as Case1) displays the inter-cluster uploads and downloads at four of the peers depicted in Fig. 1 when the *Minimal Delay Routing* algorithm is applied.

Table 1.
An example of upload and down inter-cluster load distributions in the two routing scenarios

Peer Id	Upload Case 1	Download Case 1	Upload Case 2	Download Case 2
1	0	1	1	1
2	2	1	1	1
4	2	1	1	1
5	0	1	1	1
Total	4	4	4	4

B. Fair Nodal Stress Routing Algorithm

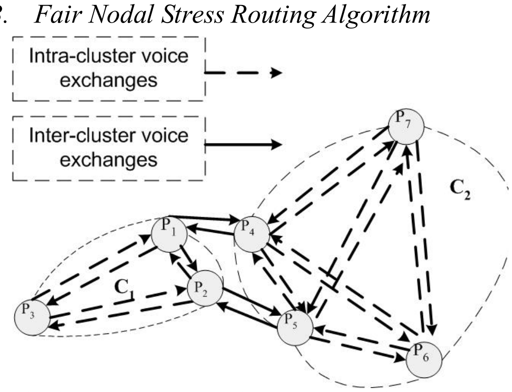

Fig. 2 Inter-cluster and intra-cluster voice exchanges and fair nodal distribution routing

In order to obtain a more even distribution of voice uploads between the peers, we have also implemented the *Fair Nodal Stress Routing* algorithm which finds paths that are not necessarily the shortest in order to minimise the difference between the number of uploads and the number of downloads at each peer. In Fig. 2, the voice streams from C_2 to P_2 are delivered along the paths starting at P_5 instead of at P_4 which

1-4244-0357-X/06/$20.00 ©2006 IEEE
This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE GLOBECOM 2006 proceedings.

is used in the shortest path routing. Such routing leads to equal number of upload and download at P_1, P_2, P_4, and P_5 as shown in the last two columns from the left in Table 1.

IV. MATHEMATICAL FORMULATION FOR CLUSTER FORMATION

Let P_i denote a particular peer where $1 \le i \le N$ (N denotes the total number of peers in the DVE). We use C_k to denote a particular cluster set up in the *Initial Cluster Formulation* where $1 \le k \le N$.

General Known Variables:

$$a_{ij} = \begin{cases} 1 \text{ if the peer } P_j \text{ can hear peer } P_i \\ 0 \text{ otherwise} \end{cases}$$

Where $j \neq i, 1 \le i \le N$

A. Initial Cluster Formation Formulation

Known variables:

d_{ij} is the distance between peer P_j and peer P_i.

δ denotes the pre-defined pair-wise distance constraint between any given pair of peers in a cluster.

$$e_{ij} = \begin{cases} 1 \text{ if } d_{ij} \le \delta \\ 0 \text{ otherwise} \end{cases} \text{ where } j \neq i, 1 \le i \le N$$

Because this formulation is decomposable to a per peer basis, for clarity, we show the case of the *kth* initial cluster formed around peer P_i in the ensuing mathematical representations.

Decision variable:

$$y_j = \begin{cases} 1 \text{ if peer } P_j \text{ is included in the } kth \text{ potential cluster} \\ 0 \text{ otherwise} \end{cases}$$

Objective Function Maximise:

$$\sum_{j=1}^{N} e_{ij} y_j \; \forall k \tag{1}$$

The objective of this formulation is to maximise the number of peers enclosed in the *kth* initial cluster.

Subject to:

$$e_{il} e_{ij} y_l y_j d_{jl} \le \delta \; \forall j, l \quad l \neq i \neq j, 1 \le l \le N \tag{2}$$

B. Final Cluster Formation Formulation

Known variables:

$$b_j^k = \begin{cases} 1 \text{ if peer } P_j \text{ is in the } kth \text{ potential cluster} \\ 0 \text{ otherwise} \end{cases}$$

$$h_j^k = \begin{cases} 1 \text{ if peer } j \text{ can hears the kth potential cluster} \\ 0 \text{ otherwise} \end{cases}$$

These two known variables are obtained as the direct results of the *Initial Cluster Formation* formulation. Please note is a peer is said to hear a cluster, if it hears at least one of the peers in that cluster.

Decision variable:

$$x_j^k = \begin{cases} 1 \text{ if peer } P_j \text{ is assigned to the } kth \text{ final cluster} \\ 0 \text{ otherwise} \end{cases}$$

$$v_j^k = \begin{cases} 1 \text{ if peer } P_j \text{ hears the } kth \text{ final cluster} \\ 0 \text{ otherwise} \end{cases}$$

u_j is a non-negative integer variable. u_j equals to the number of peers of the final cluster that peer P_j belongs to.

Objective Function Minimise:

$$\sum_{j=1}^{N} \left(u_j + \sum_{k=1}^{N} h_j^k v_j^k \right) \tag{3}$$

In this work, the average access bandwidth consumption per peer is measured by the average number of duplex (send/receive) voice exchange operations per peer. The objective of this formulation is to minimise the total number (both intra-cluster and inter-cluster) of duplex voice exchanges required to support an Immersive Communication Environment (ICE).

Subject to:

$$u_j \ge b_j^k \left(x_j^k \sum_{i=1;i\neq j}^{N} b_i^k - \sum_{i=1;i\neq j}^{N} \sum_{h=1;h\neq k}^{h=N} b_i^h x_i^h \right) \forall j,k \tag{4}$$

$$a_{ij} b_i^k x_i^k \le h_j^k v_j^k \; \forall i,j,k \tag{5}$$

$$\sum_{k=1}^{N} b_j^k x_j^k = 1 \; \forall j \tag{6}$$

V. SIMULATIONS

In our simulation experiments we use a 5000 nodes Transit-Stub topology [11] to model the Internet topology. Each peer node is assumed to be collocated with a stub router. The topology generator parameters are chosen such that the maximum propagation delay in the shortest path between two nodes is 300ms. We have run our simulations over a peer population of 50, due to the complexity of the *final cluster formation* formulation.

A. Average Bandwidth Consumption and Individual Nodal Stress

In this work, the average access bandwidth consumption per peer is measured by the average number of duplex voice exchange operations per peer. Fig. 3 shows that the average access bandwidth consumption per peer over a range of peer densities from 5 to 30. The three non-linear curves in Fig. 3 correspond to the three *pair-wise distance constraints* studied, i.e. 5, 10 and 15 meters. The linearly increasing line represents the benchmark case of full-mesh P2P (full mesh uni-cast voice exchanges for every peer).

At the same density, the average access bandwidth consumption is reduced when the *pair-wise distance constraints* are relaxed from 5 to 10 meters. As discussed in B of Section II, the clustering process achieves bandwidth reduction through replacing the bandwidth-intensive full-mesh (intra-cluster) voice exchanges with inter-cluster voice exchanges. The *pair-wise distance constraint* controls the sizes of the clusters formed. When imposing the 5 meter *pair-wise distance constraint*, the small cluster sizes lead to less bandwidth-intensive intra-cluster (full-mesh) voice exchanges but such benefit is far outweighed by the disadvantage that each inter-cluster voice exchange replaces fewer intra-cluster voice exchanges. On the other hand, the extent of reduction from 10 meters to 15 meters is much smaller than that observed from 5 meters to 10 meters. This suggests that when the *pair-wise distance constraint* is large, the advantage of having larger clusters (i.e. each inter-cluster voice exchange substituting for more intra-cluster exchanges) is less dominant over the disadvantage of having more intra-cluster voice exchanges within each cluster. Consequently, the choice of *pair-wise distance constraint* should be a balance between cluster size and the number of clusters formed for maximising access bandwidth reductions. For the scenarios studied in relation to Fig.3, 10 meters can be considered a reasonable choice.

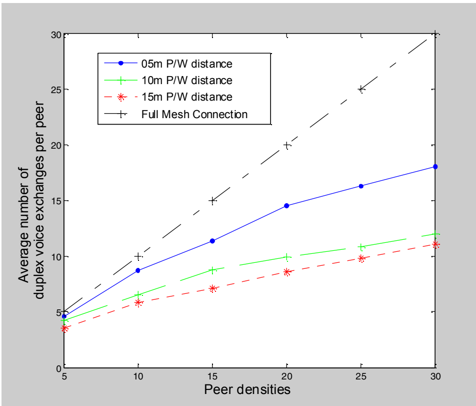

Fig. 3 Average number of duplex voice exchanges VS peer density

The three curves shown at different *pair-wise distance constraints* are all non-linear and flatten out at higher peer densities. This trend suggests that the effectiveness of clustering in bandwidth reduction is greater at higher peer densities than lower densities. In the best case scenario, at density of 30 and *pair-wise distance constraint* of 15 meters, the average bandwidth consumption is reduced by 63% from 30 streams to 11 streams. At higher peer densities, peers are closely packed in the virtual world and a reasonable number of peers can be co-located in one peer, leaving more room for improvement in bandwidth reduction by the clustering process from the full-mesh P2P scenario.

In order to study the individual nodal stress distribution between the peers, we plot the Cumulative Distribution Functions (CDFs) of uploads and downloads at each peer for three interesting scenarios, shown respectively on Figures 4, 5 and 6 (see next page). At the density of 30, the comparison between Figures 4 and 5 shows that when the *Minimal Delay Routing Algorithm* is implemented, the distribution of individual nodal stresses is much more even or fairer in download than in upload. The distribution of individual nodal stresses in download is unaffected by the choice of routing algorithms as each peer always receives one stream from each of the clusters heard by that peer. Figure 6 shows the distribution of individual nodal stresses in uploads at the density of 30 when the *Fair Nodal Stress Routing Algorithm* is applied. The comparison between Fig. 5 and Fig. 6 illustrates that the *Fair Nodal Stress Routing Algorithm* yields a much more even distribution of individual upload stresses than the *Minimal Delay Routing Algorithm*.

B. *Peer to Peer Delays*

Table 2 shows, over different densities and different *pair-wise distance constraints*, the average (per pairs of communication peers) percentage increase in delay over the full-mesh P2P case when the *Minimal Delay Routing Algorithm* is applied in conjunction with the two clustering models. The two hop routes found by the *Minimal Delay Routing Algorithm* for inter-cluster voice exchanges account for the increases in delay over the best-delay scenario of full-mesh P2P. In the worst case scenario, at density of 25 and *pair-wise distance constraint* of 15 meters, the average percentage increase in delay is 31.51% (actual average delay 179 milliseconds, average full mesh delay 140 millisecond). Table 3 shows the difference between the percentage increase in delay incurred by Minimal Delay Routing and the percentage increase in delay incurred by Fair Nodal Stress Routing. When applying the Fair Nodal Stress Routing Algorithm instead of the *Minimal Delay Routing Algorithm*, the extra delay penalty incurred is reasonably small. In the worst case of, at density of 25 and *pair-wise distance constraint* of 15 meters, the application of Fair Nodal Stress Routing incurs an additional 7.62% increase in delay over the full-mesh P2P case, on top of the 26.21% incurred when Minimal Delay Routing is applied.

C. *Peer to Peer Directional Error*

We measured the average (per pairs of communication peers) *percentage directional error* over different peer densities and pair-wise distance constraints. The *percentage directional error* is the percentage difference between the rendering *centroid* (see B of Section II) and the exact positions of the speaking peers with respect to the position of the listening pee. In the worst case scenario of peer density of 30 and pair-wise distance set to 15m, 86% of pairs of listening and speaking peers have zero directional errors. For the next ten percentiles (up to 96%), the directional error is below 141%. The next 4 percentiles (to 100%), the error is below 501.33% error. The relevant tables and graphs are omitted due to space limits.

Table 2.
Delay percentage increase Minimal Delay Routing

P/w distance	Density Of 5	Density Of 10	Density Of 15	Density Of 20	Density Of 25	Density Of 30
5 m	3.23%	2.33 %	11.40%	11.99%	19.62%	19.81%
10 m	8.06%	13.40%	20.86%	21.24%	28.18%	27.62%
15 m	17.03%	16.26%	25.81%	23.67%	31.51%	26.21%

Table 3.
Extra delay percentage increase Fair Nodal Distribution Routing

P/w distance	Density Of 5	Density Of 10	Density Of 15	Density Of 20	Density Of 25	Density Of 30
5 m	0 %	0 %	4.05 %	6.33 %	5.11 %	5.04 %
10 m	4.32 %	5.67 %	6.02 %	4.99 %	6.33 %	6.63 %
15 m	4.44 %	5.01 %	7.07 %	5.49 %	5.20 %	7.62 %

D. *Peer to Peer Attenuation Error*

The attenuation error measured in our simulations applies to the rendering of inter-cluster streams. Our attenuation error metric is defined as the difference between the chosen attenuation distance for a mixed stream and the exact distance of an individual stream, normalised as a percentage of the hearing range. We measured the Cumulative Distribution

Function of the attenuation errors in the worst case scenario, at peer density of 30 and *pair-wise distance* of 15 m. In that scenario, the maximal error is 47% of hearing range, and 90% of pairs of peers have error less than 30% of the hearing range. The relevant tables and graphs have been omitted due to the space constraints.

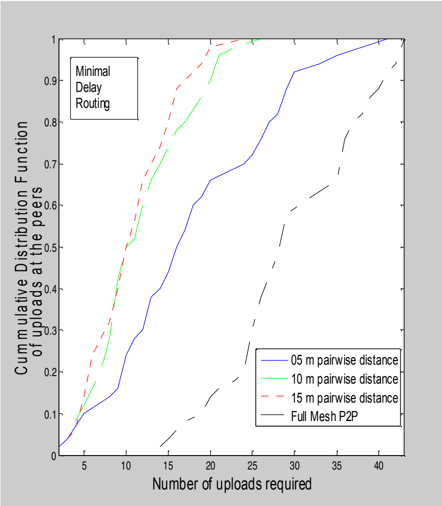

Fig. 4 Uploads CDFs at peer density of 30, Minimal Delay Routing

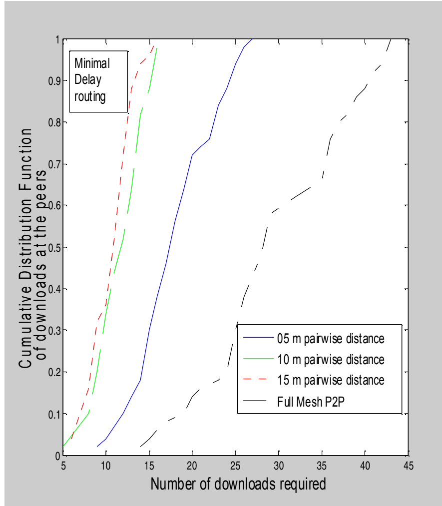

Fig. 5 Downloads CDFs at peer density of 30, Minimal Delay Routing

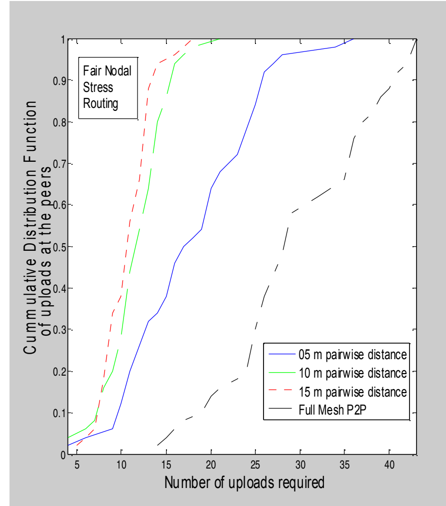

Fig. 6 Uploads CDFs of at peer density of 30, Fair Nodal Stress Routing

VI. CONCLUSION

In this paper, we proposed a Peer-to-Peer (P2P) architecture to deliver a multi-party immersive voice communication service, called Immersive Communication Environment (ICE) for users accessing a Distributed Virtual Environment (DVE). In comparison to the "brute force" full-mesh P2P approach, real-time voice deliveries in our architecture demands lower access bandwidth at the peers. This bandwidth reduction is brought about by the clustering of pair-wise close peers and the mixing of the voice streams within the same cluster which reduces the number of direct peer to peer voice exchanges. This bandwidth reduction is improved by the relaxing of the pair-wise distance constraint and is larger at larger peer densities. In the best case scenario, the average bandwidth consumption is reduced by 63% from 30 streams to 11 streams. We implement full-mesh uni-cast voice deliveries between the peers within the same cluster in order to ensure high voice quality in delay and rendering for these close-by peers. For the mixed voice streams exchanged between clusters, our architecture minimises the rendering error in those voice streams by choosing an appropriate directional *centroid* and the appropriate attenuation distance. Our simulation results show that the directional error and attenuation error for most voice streams are reasonably small. To minimise the delay penalties incurred in inter-cluster voice exchanges, we at first implemented the *Minimal Delay Routing algorithm* which finds the shortest path between peers across clusters. However this *Minimal Delay Routing* algorithm causes uneven bandwidth consumptions at many peers. The *Fair Nodal Stress Routing algorithm* is thus implemented to find sub-minimal paths that contribute to a more even distribution of uploads among the peers. Our simulations find that the *Fair Nodal Stress Routing algorithm* incurs maximally an additional 7.62% increase in delay over the full-mesh P2P case, on top of the 26.21% incurred when Minimal Delay Routing is applied. In comparison to the full-mesh P2P, our P2P architecture delivers real-time immersive voice streams with a good balance between scalability access bandwidth consumptions at the peers, and the rendering and delay qualities of voice streams.

ACKNOWLEDGEMENT

This work is supported by the Co-operative Research Centre for Smart Internet CRC (SITCRC) and the University of Wollongong (UOW), Australia.

REFERENCES

[1] Paul Boustead and Farzad Safaei, "Comparison of Delivery Architectures for Immersive Audio in Crowded Networked Games", in *Proc. of the 14th ACM International Workshop on Network and Operating Systems Support for Digital Audio and Video (NOSSDAV)*, Ireland, 16-18 June, 2004.

[2] J. Bolot and F. Parisis, "Adding Voice to Distributed Games on the Internet", in *Proc. of the Seventeenth Annual Joint Conference of the IEEE Computer and Communications Societies,* pages 480-487, 1998.

[3] M. Radenkovic, C. Greenhalgh, S. Benford, "Deployment issues for multi-user audio support in CVEs," in *ACM Symposium on Virtual Reality Software and Technology*, 2002, pp. 179-185.

[4] MMOGCHART.com, http://www.mmogchart.com/ (9 Aug. 2005).

[5] SiderWinder Game Voice, htttp://www.gamevoice.com , (12 Mar. 2006).

[6] Skype, http://www.skype.com, (12 Mar. 2006).

[7] TeamSpeak, The Team Pay Engine, http://www.teamspeak.org, (9 Aug. 2005).

[8] Ying Peng Que, Paul Boustead, and Farzad Safaei, "Minimising the Computational Cost of Providing a Mobile Immersive Communication Environment (MICE)", in *Proc. of the 2nd IEEE International Workshop on Networking Issues in Multimedia Entertainment (NIME 06) at Consumer Communications and Networking Conference (CCNC)*, Las Vegas, Nevada, U.S.A, 7-10 Jan. 2006.

[9] C. D. Nguyen, F. Safaei, P. Boustead, "A distributed server architecture for providing immersive audio communication to massively multi-player online games," in *Proc. of IEEE International Conference on Networks (ICON)*, Singapore, November 2004, pp. 170-176.

[10] Durand R. Begault, *3-D Sound for Virtual Reality and Multimedia*, Academic Press Professional, Cambridge, MA, USA, 1994.

[11] E.W. Zegura, K. Calvert, and S. Bhattacharjee. How to Model an Internetwork. INFOCOM 1996.